The following is claimed:

1. A method for coding of a speech signal, the method comprising the steps of:

accumulating samples of the speech signal over at least a minimum sampling duration;

evaluating the accumulated samples associated with the minimum sampling period to obtain a representative sample;

determining whether a slope of the representative sample of the speech signal conforms to a defined characteristic slope stored in a reference database of spectral characteristics; and

selecting a value of a coding parameter for application to the speech signal for the coding based on the determination on the spectral slope of the representative sample.

2. The method according to claim 1 where the selecting comprises selecting a first coding parameter value as the value if the representative sample of the speech signal is sloped in accordance with the defined characteristic slope.

3. The method according to claim 1 where the selecting comprises selecting a second coding parameter value as the value if a slope of the representative sample of the speech signal is generally flat in accordance with the determining step.

4. The method according to claim 1 where the evaluating comprises averaging the accumulated samples over the minimum sampling duration to obtain the representative sample.

5. The method according to claim 1 further comprising the step of assuming the spectral response of a speech signal is sloped in accordance with the defined characteristic slope prior to completion of at least one of the accumulating step and the determining step.

6. The method according to claim 1 wherein the selecting step comprises selecting a first coding parameter value as the value of an initial default coding parameter based on the assumption that the spectral response of the speech signal is sloped in accordance with the defined characteristic slope.

7. The method according to claim 1 where the defined characteristic slope approximately represents a Modified Intermediate Response System.

8. The method according to claim 1 wherein the selecting comprises selecting at least one preferential encoding parameter value as the value; an encoding parameter underlying the at least one preferential encoding parameter value and including one or more of the following: pitch gain per frame or subframe, at least one filter coefficient of a perceptual weighting filter, at least one bandwidth expansion constant associated with a synthesis filter, and at least one bandwidth expansion constant associated with an analysis filter.

9. The method according to claim 1 where the selecting comprises selecting at least one preferential decoding parameter value as the value; a decoding parameter underlying at least one decoding parameter value and including one or more of the following: at least one bandwidth expansion constant associated with a synthesis filter and at least one linear predictive filter coefficient associated with a post filter.

10. The method according to claim 1 where the selecting comprises adjusting the value of a coding parameter selected from the group consisting of pitch gains per

frame or subframe, at least one filter coefficient of a perceptual weighting filter, at least one bandwidth expansion constant associated with a synthesis filter, at least one bandwidth expansion constant associated with an analysis filter, and at least one linear predictive filter coefficient associated with a post filter.

11. The method according to claim 1 further comprising adjusting a bandwidth expansion of the speech signal as the value for at least one of a synthesis filter and an analysis filter from a previous value to a revised value based on a degree of slope or flatness in the speech signal.

12. The method according to claim 1 where the selecting comprises selecting a bandwidth expansion value of the speech signal as the value in conformance with the following equations:

$$\frac{1}{A(z)} = \frac{1}{1-\sum_{i=1}^{P} a_{i\,\mathrm{revised}} Z^{-1}},$$

where $1/A(z)$ is a filter response represented by a z transfer function, ai revised is a linear predictive coefficient, $i = 1 \ldots P$, and P is the prediction order or filter order of the synthesis filter,

$$a_{i\,\mathrm{revised}} = a_{i\,\mathrm{previous}} \gamma^{i},$$

where $a_{i\,\mathrm{revised}}$ is a revised linear predictive coefficient, $a_{i\,\mathrm{previous}}$ is a previous linear predictive coefficient, γ is the bandwidth expansion constant, $i = 1 \ldots P$, and P is the prediction order of the synthesis filter of the encoder, and where $a_{i\,\mathrm{previous}}$ represents a member of the set of extracted linear predictive coefficients $\{a_{i\,\mathrm{previous}}\}^{P}{}_{i=1}$, for the synthesis filter of the encoder.

13. The method according to claim 12 where the value of the bandwidth expansion constant for a generally flat spectral response differs from that of the defined characteristic slope.

14. The method according to claim 12 where the value of the bandwidth expansion constant is greater for a generally flat spectral response than the defined characteristic slope.

15. The method according to claim 12 where γ is set to a first value of approximately .99 if the slope of the representative sample is consistent with an MIRS spectral response and γ is set to a second value of approximately .995 where the slope of the representative sample is generally flat or approaches zero.

16. The method according to claim 1 wherein the selecting comprises selecting a frequency response factor of a perceptual weighting filter as the value of the coding parameter based on a degree of slope or flatness in the speech signal.

17. The method according to claim 1 further comprising controlling a frequency response of a perceptual weighting filter based on the following equation:

$$W(z) = \frac{1}{1-\alpha z^{-1}} \frac{1+\sum_{i=1}^{P} a_i \rho^i z^{-i}}{1+\sum_{i=1}^{P} a_i \beta^i z^{-i}}$$

where α is a weighting constant as the value of the coding parameter, β and ρ are preset coefficients, P is the predictive order, and $\{a_i\}$ is the linear predictive coding coefficient.

18. The method according to claim 17 wherein the controlling comprises selecting different values of the weighting constant α to adjust the frequency response of the perceptual weighting filter in response to the determined slope or flatness of the speech signal.

19. The method according to claim 17 further comprising controlling the value of α based on the spectral response of the speech signal such that α

approximately equals .2 where the speech signal is consistent with the MIRS spectral response and α approximately equals 0 where the speech signal is consistent with a generally flat signal response.

20. The method according to claim 1 further comprising the step of selecting a frequency response factor of a post filter as the value of the coding parameter based on a degree of slope or flatness of the speech signal.

21. The method according to claim 1 further comprising the step of controlling a frequency response of a post filter in accordance with the following equation:

$$P(z) = \frac{1 + \sum_{i=1}^{P} a_i \gamma_1^i z^{-i}}{1 + \sum_{i=1}^{P} a_i \gamma_2^i z^{-i}}$$

where γ_1 and γ_2 represents a set of post-filtering weighting constants in which the value is a member of the set, {ai} is the linear predictive coding coefficient, and P is the filter order of the post filter.

22. The method according to claim 21 further comprising the step of controlling a frequency response of a post filter by selecting different values of post-filtering weighting constants of γ_1 and γ_2 in response to the determined slope or flatness of the speech signal.

23. The method according to claim 21 where γ_1 and γ_2 approximately equal .65 and .4, respectively, if the speech signal is consistent with an MIRS spectral response; and where γ_1 and γ_2 approximately equal .63 and .4, respectively, if the speech signal is consistent with a generally flat signal response.

24. A system for coding a speech signal, the system comprising:

a buffer memory for accumulating samples of the speech signal over at least a minimum sampling duration;

an averaging unit for evaluating the accumulated samples associated with the minimum sampling period to obtain a representative sample;

a storage device adapted to store spectral characteristics for classifying the speech signal as a closest one of a defined characteristic slope and a flat speech signal;

an evaluator adapted to determine whether a slope of the representative sample of the speech signal conforms to a defined characteristic slope stored in the storage device; and

a selector for selecting a preferential one of a first coding parameter value and a second coding parameter value for application to the speech signal for the coding based on the determination on the slope of the representative sample.

25. The system according to claim 24 where the selector is adapted to select the first coding parameter value as the preferential one if the evaluator determines that a slope of the representative sample of the speech signal generally conforms to the defined characteristic slope.

26. The system according to claim 24 where the selector is adapted to select the second coding parameter value as the preferential one if the evaluator determines that a slope of the representative sample of the speech signal is generally flat.

27. The system according to claim 24 where the evaluator comprises an averaging unit is adapted to average the accumulated samples over the minimum sampling duration to obtain the representative sample.

28. The system according to claim 24 where the evaluator assumes the spectral response of a speech signal is sloped in accordance with the defined characteristic slope prior to the expiration of the minimum sampling duration.

29. The system according to claim 24 where the defined characteristic slope approximately represents a Modified Intermediate Response System.

30. The system according to claim 24 where the evaluator triggers an adjustment of at least one encoding parameter to a revised encoding parameter during the coding process.

31. The system according to claim 24 where the evaluator is coupled to a coder, where the evaluator sends at least one of a control data and a spectral-content indicator to the coder for controlling one or more of the following coding parameters: (a) pitch gains per frame or subframe, (b) at least one filter coefficient of a perceptual weighting filter of an encoder, (c) at least one filter coefficient of a synthesis filter of an encoder, (d) at least one bandwidth expansion constant associated with a synthesis filter of the coder, (e) at least one bandwidth expansion constant associated with a synthesis filter of a decoder, (f) at least one bandwidth expansion constant associated with an analysis filter of an encoder, and (g) at least one filtering coefficient associated with a post filter coupled to a decoder.